

THE G4BWE audio filter

by STEPHEN PRICE, G4BWE*

ONE of the most important components contained within an ssb transceiver or communications receiver is the intermediate frequency filter. Apart from playing a vital role in the generation of ssb for modern transceivers, it is the i.f. filter which dictates the ultimate bandwidth of the receiver. The filter's characteristics will normally be tailored to provide an audio spectrum at the receiver's output extending over the range 300Hz to approximately 3kHz. Such a relatively wide bandwidth is essential to capture the complex harmonic structure of speech waveforms, and any significant reduction in this bandwidth will considerably impair the intelligibility of phone signals. In contrast, morse telegraphy (cw) transmissions contain only a small amount of information/unit time, and for these signals the optimum receive bandwidth is very much narrower.

If the amateur frequency allocations were held subject to a divine plan and so contained only a finite number of strong, highly stable transmissions, each well spaced from its neighbours and devoid of all man-made and atmospheric interference, such considerations of receiver bandwidth would become largely academic. In reality, however, the electromagnetic spectrum is a veritable jungle, densely populated with a myriad of randomly-spaced signals and various kinds of interference. A rational solution to the problem of achieving adequate reception on crowded and noisy amateur bands is to provide some means of tailoring the receiver's bandwidth so that it matches the information rate of the transmission.

For cw reception it is possible to employ a specialized i.f. filter with a bandwidth of only a few hundred hertz, which can be switched into operation when required. Unfortunately, cw filters of this type tend to be rather expensive and are inherently inflexible as their bandwidth and centre frequency cannot be readily altered. A very practical alternative to the cw i.f. filter is an active audio filter. Such filters, when carefully designed, provide a high degree of flexibility, as it is possible to adjust both bandwidth and centre frequency by the manipulation of separate controls. In consequence a host of designs for audio filters have appeared during the last few decades, and many offer considerable advantages. It is interesting to note that an audio filter can be so engineered that it will provide not only a bandpass characteristic but also a bandstop or "notch" function, thus providing a means of eliminating major interference occurring at a specific frequency. This alternative to the technique of merely reducing overall bandwidth is particularly useful in enhancing the quality of ssb reception, as stray heterodynes can be readily eliminated without significantly impairing voice intelligibility.

The graphic equalizer

Turning for a moment to the world of hi-fi and professional sound recording, one finds that an entirely new type of frequency equalizer or, to employ more colloquial terminology, "tone control", has recently emerged. An older type of tone control network, developed by Peter Baxandall, features only two potentiometers labelled BASS and TREBLE. These controls may be adjusted to provide deviation from a flat frequency response so that gain (boost) or, alternatively, attenuation (cut) can be achieved at either the low or high frequency end of the audio spectrum respectively.

The newcomer, known as the "graphic equalizer", is considerably more complex, and consists of a large number of separate filter networks, each with a relatively narrow bandwidth. Slider-type potentiometers are employed to control these networks, so that varying degrees of boost or cut may be obtained at specific points within the audio spectrum. Fig 1 demonstrates both pictorially and diagrammatically how the graphic equalizer operates, and shows the visual, or "graphic" presentation of the response characteristic afforded by the use of slider rather than rotary type controls. The professional graphic equalizer is designed to function over the entire audible spectrum, ie from 30Hz to 20kHz, and each slider control will provide a maximum of around 15dB of either boost or cut at its resonant frequency.

The author decided to build a specialized form of graphic equalizer for communications use with the following differences: (a) the design covers a more limited frequency band of 300Hz to 3kHz, and (b) each slider control offers a much greater range of amplitude control.

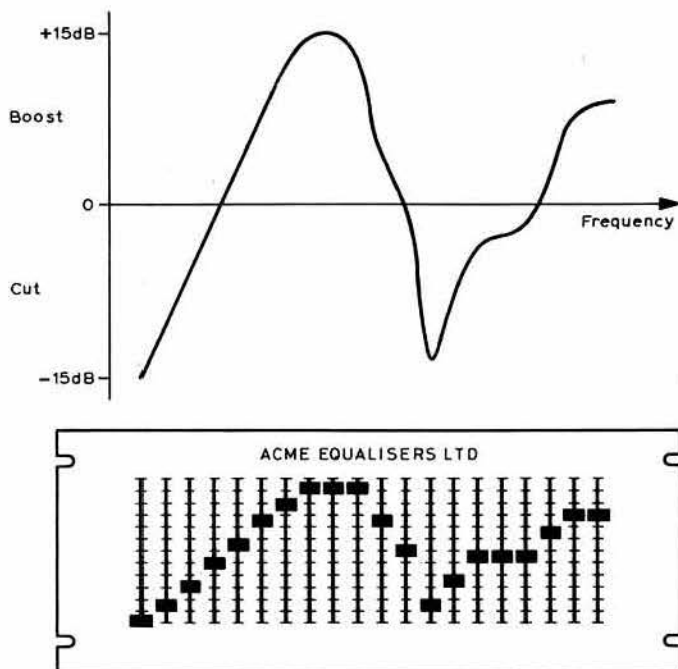


Fig 1. The graphic equalizer

How the filter works

Before examining the complete design in detail it will be useful to study the principle of operation. Fig 2 shows a skeleton circuit diagram of the filter. The series-tuned circuit consisting of L and C exhibits a low impedance at its resonant frequency. If the potentiometer wiper is set at point A, resistor R_1 , in conjunction with the tuned circuit, forms a potential divider which will attenuate the signal being presented to the non-inverting input of the operational amplifier. The result of this action is to produce a notch characteristic centred on the resonant frequency of L and C. Conversely, if the wiper is positioned at point B, resistor R_F forms a second potential divider in conjunction with the same tuned circuit. This divider serves to attenuate the negative feedback voltage presented to the inverting input of the op-amp, and thus increases the amplifier's voltage gain considerably. A bandpass or "boost" characteristic, once again centred on the tuned circuit's resonant frequency, has therefore been obtained.

If R_1 and R_F are made identical in value and the potentiometer is set at its midpoint, the input attenuation becomes equal to the amplifier's gain, and the gain of the complete circuit is therefore unity. Moving the potentiometer's wiper to either side of centre will enable varying degrees of boost (peak) and cut (notch) to be introduced. As explained earlier, the equalizers employed in recording studios provide only limited amounts of boost and cut. This restriction in control range may be provided by inserting fixed resistors in series with each end of the potentiometer's track. The author's filter dispenses with these resistors and thereby enables the operator to achieve the highest possible ratio in level between wanted and unwanted frequencies.

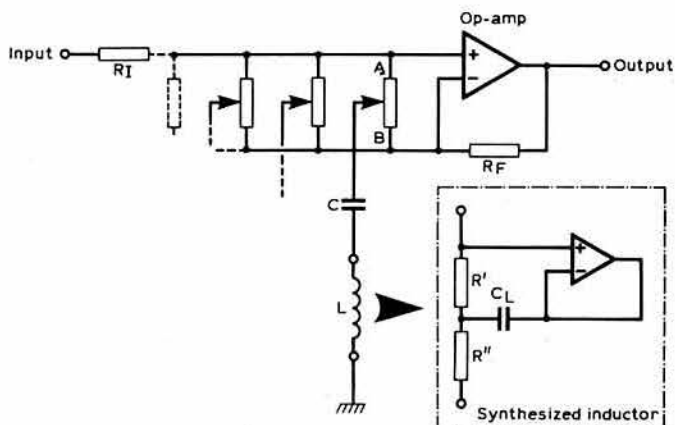


Fig 2. Skeleton circuit showing the filter's principle of operation

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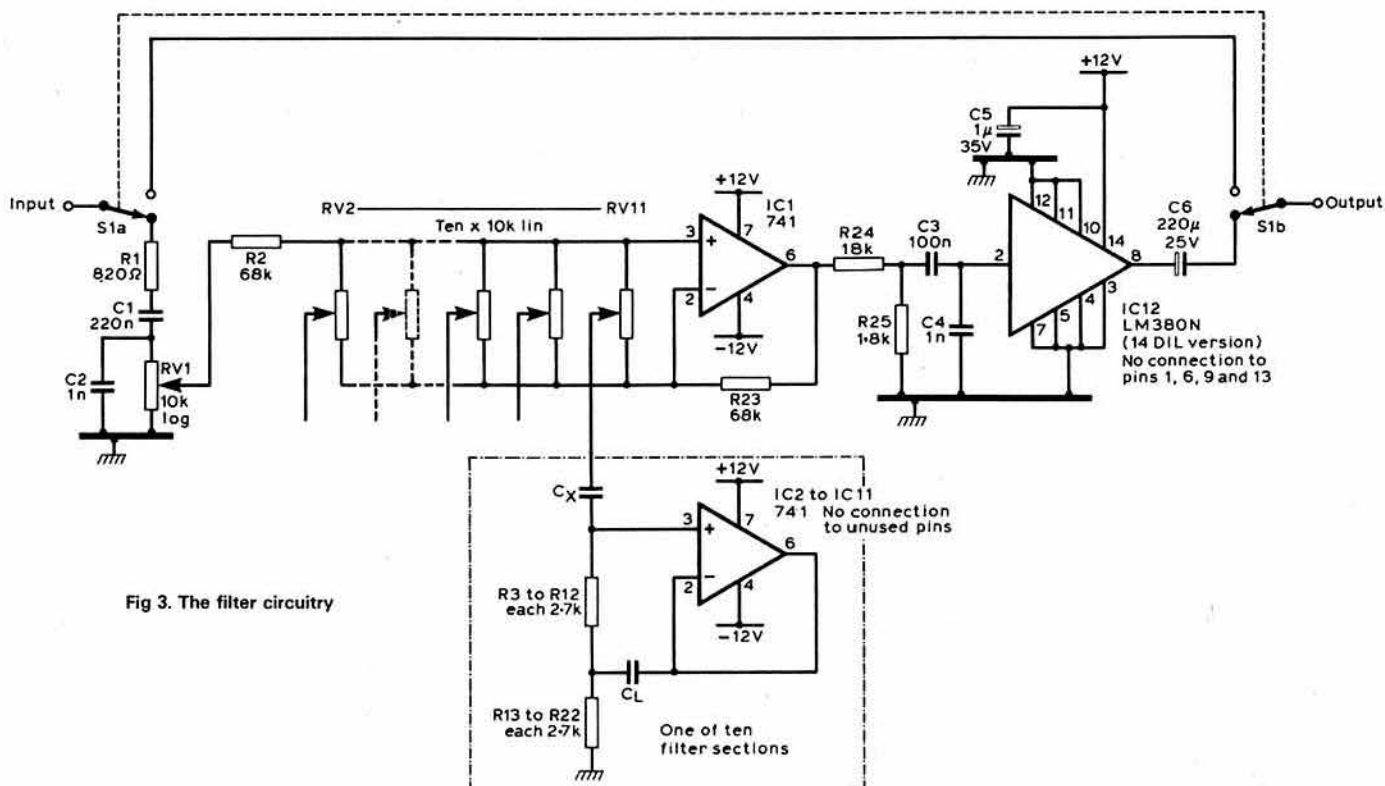


Fig 3. The filter circuitry

Rather than employing discrete inductors, which are bulky, expensive and susceptible to hum pick-up, the inductance required for each tuned circuit is synthesized by a simple gyrator consisting of an inexpensive op-amp, two fixed resistors and a capacitor (see box, Fig 2). For those interested, the value of synthesized inductance in Henries is given by the following formula:

$$L = C_L \times R' \times R''$$

Many previous audio filter designs, particularly those presented in the more theoretically-based textbooks and articles, have suffered from the disadvantage of requiring accurate component values, thus necessitating the use of close tolerance capacitors which are expensive and difficult to obtain. The design and operational parameters of the author's filter have been framed to allow the use of standard five per cent tolerance resistors and five or ten per cent tolerance non-electrolytic capacitors. The final design utilizes 10 filter sections, with centre frequencies extending over the range 440 to 2,720Hz. The first five sections each feature a bandwidth of 200Hz and thereby cover the segment 350 to 1,360Hz. The next three sections are of 250Hz (-6dB) bandwidth, and the ninth and tenth sections have bandwidths of 300 and 400Hz respectively. It is of course possible to design a filter with sections of 200Hz bandwidth only, but 13 separate filter networks are required to cover the range 350 to 2,920Hz; three more than the number employed in the present design. Furthermore, as the centre frequency of a filter of given bandwidth is increased, the effect of component tolerances becomes proportionately greater. For example, a filter of

200Hz bandwidth resonant at 2,820Hz would demand fixed component tolerances of around 2.5 per cent or, alternatively, the incorporation of variable elements in order to set the centre frequency accurately.

The practical design

Fig 3 shows the audio filter circuitry. The input signal obtained from the rig's external loudspeaker socket is fed via the by-pass switch (S1a), through R1 and the coupling capacitor C1, to the gain control, RV1. C2 removes any high frequency energy and acts as an rf filter.

Reference to Fig 2 and associated text will explain the function of R2, R23, RV2 to RV11 and op-amp IC1. For the sake of clarity, only one of the 10 filter sections is shown. Each section is identical in circuitry, the only variation being in the values of capacitors CX and CL. Table 1 gives these values plus other relevant data. The output from pin 6 of IC1 is fed via the potential divider comprising R24 and R25 to coupling capacitor C3, and hence to the input pin of IC12. IC12 is a monolithic audio power amplifier type LM380N, and inclusion of this final stage enables the filter to drive a loudspeaker or low impedance headphones. C6 is the output coupling capacitor and S1b completes the by-pass circuitry. Decoupling capacitor C5 is critical in that it serves to prevent oscillation of IC12. C5 should therefore be positioned close to IC12 and must be a low-inductance type, eg tantalum bead. C4 decouples the input of IC12 at high frequencies.

The mains power supply circuitry is shown in Fig 4 and requires little explanation. As there are a large number of op-amps in the filter, 11 in total,

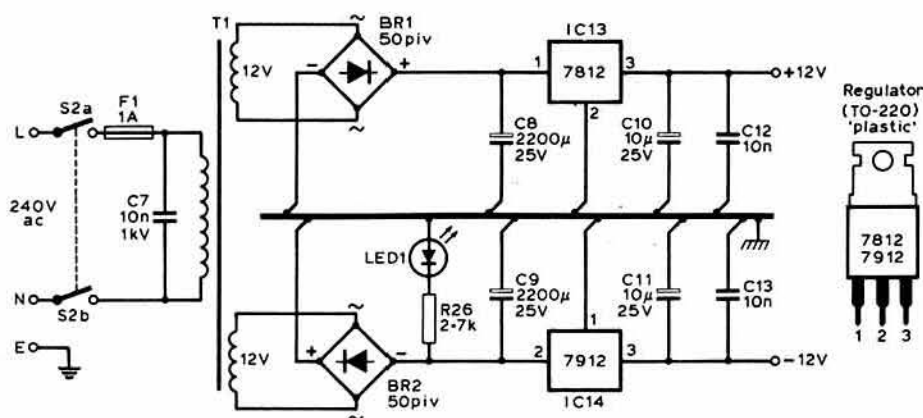


Fig 4. Mains psu for the filter

the author opted for a two-rail supply. Small 1A ic regulators are employed, and readers should note the variation in pin-out between the positive output (7812) and negative output (7912) devices. LED 1, which is fed via current limiting resistor R26, provides on/off indication. The mains transformer is a small general purpose type with 250mA (minimum) secondaries, as manufactured by Douglas, Repanco etc.

Construction and testing

As the filter is an audio project the exact layout of components becomes largely a matter of choice, but hum loops should be avoided by the use of single-point earthing. It is also a good idea to employ miniature audio coaxial cable for the signal leads. The prototype has been constructed on Veroboard, and the 10 slider potentiometers are bolted directly to the filter's case. Readers who demand a more professional standard of construction may wish to consider the production of a pcb, and if this technique is adopted it should be possible to mount the slider potentiometers on the circuit board itself, thus cutting down on the number of flying leads. A metal case is recommended, mainly as a precaution against the effects of transmitter radiation. The most tricky part of the filter's construction is undoubtedly the cutting of the 10 slots for the slider controls. However, a small twist drill, a supply of junior hacksaw blades and a miniature file should make light work of the job.

Two sockets are provided on the back panel for input and output. These may be either jack, phono or DIN according to preference. The only other panel-mounted components are S1 (by-pass), S2 (on/off), RV1 (gain) and LED1.

Op-amps IC1 to IC11 may be mounted in sockets, provided these are of proven quality (eg Texas low profile). The LM380N must be soldered into place, as this is a power device and heat is conducted away from the chip via pins 3, 4, 5, 10, 11 and 12. The Vero strips to which these pins are

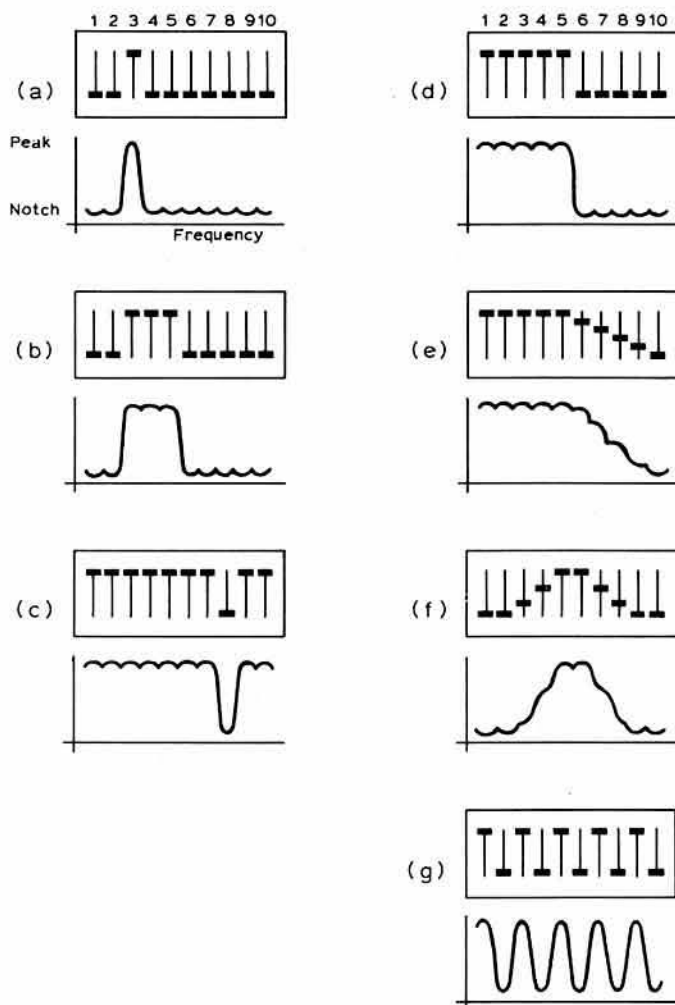


Fig 5. Some response curves obtainable. (a) A narrow bandwidth cw filter, centre frequency 860Hz. (b) Wide-bandwidth cw filter, centre frequency 1kHz. (c) A 2kHz notch. (d) Steep-slope low-pass filter, turnover frequency 1,400Hz. (e) Gentle-slope low-pass filter. (f) Low-Q band-pass response centred on 1,400Hz. (g) Comb filter, see text

Table 1. Values of C_X and C_L plus other relevant data

Section	C_X	C_L	Centre Frequency (Hz)	Bandwidth (Hz)	Q	Synthesized Inductance (H)
1	18nF	1 μ F	440	200	3.7	7.3
2	8.2nF	1 μ F	650	200	5.5	7.3
3	4.7nF	1 μ F	860	200	7.3	7.3
4	3.3nF	1 μ F	1030	200	8.7	7.3
5	2.2nF	1 μ F	1260	200	10.7	7.3
6	1.8nF	820nF	1530	250	10.7	6
7	1.3nF*	820nF	1800	250	12.6	6
8	1nF	820nF	2060	250	14.3	6
9	940pF**	680nF	2330	300	13.4	5
10	1nF	470nF	2720	400	10.8	3.4

*1.2nF + 100pF in parallel **470pF + 470pF in parallel

soldered should therefore be left uncut for a length of approximately 1.5in on either side of the LM380N and liberally tinned with solder. If a pcb is employed, an area of unetched copper should be left as a heatsink.

Suitable 10 per cent tolerance capacitors for C_L (see Table 1) are Mullard C280 polyester (352 series). It may not be possible to obtain the 820nF value, however, but use of 470nF in parallel with 330nF provides an acceptable substitute. The C280 series capacitors are not suitable for C_X , as below 330nF their stated tolerance is increased to 20 per cent. Siemens polystyrene type B31310 of five per cent tolerance, or the ceramic plate capacitors sold by Marshalls, may be used for C_X . A strict analysis of the design will show that the use of 5 or 10 per cent tolerance capacitors for C_L and C_X does not guarantee a particularly high accuracy in the setting of centre frequencies for the filter sections that operate above 1kHz. No problems have been encountered in practice, however, and in any case it is a simple matter to trim individual values by the addition of padding capacitors should any of the sections fail to operate as intended.

The completed filter may be tested with the aid of an audio frequency generator and ac millivoltmeter or, alternatively, by the following procedure:

The filter's input is coupled to the receiver's external loudspeaker socket, and a loudspeaker is connected to the filter's output. The receiver is adjusted to provide a broad spectrum of random noise, ie a fairly constant "hiss" or "roar" which can be obtained by tuning the receiver to a clear frequency or by simply unplugging its antenna and turning all the gain controls, including af, to maximum. The by-pass switch (S1) is set to bring the filter into operation, and with every one of the 10 sliders in the notch position and RV1 at mid-travel a low output level should result.

The first slider (section 1, 440Hz) is now advanced to the peak position, and if all is working properly the filter will deliver a significantly higher noise level. This noise will possess a distinct tonality or "pitch". The first slider is now returned to the notch position and the second slider advanced to peak. Once again, a distinct noise output is obtained but the "pitch" will appear noticeably higher due to the upward shift in resonant frequency. This process is repeated so that all 10 sections are sequentially "peaked" and, provided the noise pitch continues to rise in the expected manner, the filter may be considered ready for use.

Fig 5 presents a few examples of the response curves obtainable, and serves to illustrate the filter's versatility. The comb response, example g, is of little practical value, but if the receiver's calibrator tone is swept through the range 300Hz to 3kHz five distinct peaks accompanied by alternate notches should be clearly heard. This exercise provides an amusing demonstration of the filter's capabilities.

When using the filter to provide a narrow passband for cw reception, as in Fig 5 example (a), it should be remembered that not all present-day cw transmissions are free from chirp. Chirpy cw signals will sweep in and out of a narrow passband, thus making copy virtually impossible. The only solution to this problem will be to widen the passband as shown in example (b).

Refinements

Modern ssb receivers and transceivers are characterized by having an audio response curve which rolls-off fairly rapidly below 300Hz. There are two main reasons for this. First, the i.f. response of a transceiver will be tailored to provide maximum carrier rejection on transmit, and it is this passband shaping that results in a restricted lf response on receive. Second, the use of physically small "communications" loudspeakers, invariably housed in vented enclosures, produces considerable attenuation of the lower audio frequencies. In most cases, therefore, the 10-section design featuring a first filter resonance of 440Hz will prove adequate. However, readers who possess a receiver system which features extended lf response may wish to incorporate an additional section with a centre frequency of 230Hz. The values of C_L and C_X for this section are 1 μ F and 68nF respectively.

Components list

R1	820Ω	C1	220nF
R2, 23	68kΩ	C2, 4	1nF
R3-22, 26	2.7kΩ	C3	100nF
R24	18kΩ	C5	1μF tant elect 35V
R25	1.8kΩ	C6	220μF elect 25V
All 0.25W 5%		C7	10nF 1kV
RV1	10kΩ log	C8, 9	2,200μF elect 25V
RV2-11	10kΩ linear slider	C10, 11	10μF elect 25V
LED1	TIL209 (red)	C12, 13	10nF
BR1, 2	50 pV silicon bridge rectifiers	C _L , C _X	see Table 1 and text
S1, 3	DPDT toggle	IC1-IC11	741
S2	DPST mains	IC12	LM380N 14 dil
F1	1A anti-surge	IC13	7812 regulator
T1	240V primary, 0-12V, 0-12V secondaries 250mA	IC14	7912 regulator

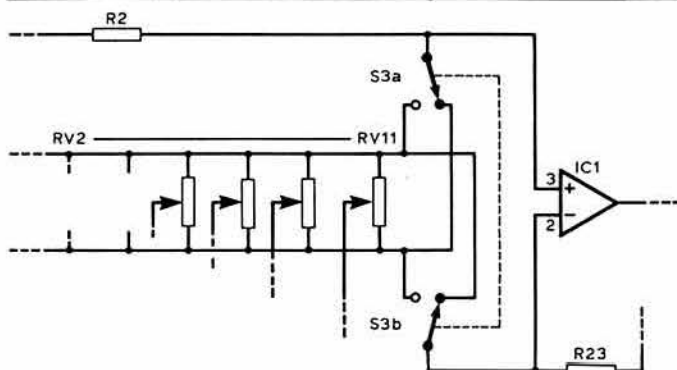


Fig 6. Circuit modifications to incorporate S3

A valid criticism of the basic design is that the need to manipulate 10 entirely separate slider potentiometers can, at times, make operation of the filter a rather tiresome process. The problem is largely overcome by the addition of a single switch (see Fig 6 for circuit modifications) that enables the amplitude response to be inverted. S3, if incorporated, may therefore

be employed to instantly convert a given bandpass response into a band-stop (notch) characteristic. Furthermore, a low-pass response may be transformed, by simply flicking S3, into a complementary high-pass response. In order to avoid confusion, S3 must be clearly labelled **NORMAL** and **INVERT**.

Finally, constructors may wish to add a second output socket specifically for headphones. If modern low impedance headphones are to be used, a fixed resistor of around 150Ω should be wired in series with this socket to attenuate the filter's output, thereby reducing the effect of background noise generated primarily by IC1. While considering the noise generated by IC1, constructors may note that it is worthwhile selecting the lowest-noise 741 from those purchased and to use it as IC1. The remaining ics, to be used in the gyrators, will not contribute significant noise.

Conclusion

Whether used in conjunction with a simple direct-conversion receiver or a highly-sophisticated multimode transceiver, the filter described will more often than not provide a valuable improvement in the readability of weak cw signals. It will also enable the operator to remove annoying heterodynes and other "hash" from telephony transmissions.

As the process of separating wanted from unwanted information is also a major function of the human brain, the filter can be said to be working in harmony with the operator's sensory and "processing" systems. In future, however, automation will play an ever-increasing role in information processing. The morse decoder, whether built around committed logic or microprocessor based, is a perfect example of what technology holds in store. It must be remembered, however, that these machines, still in their infancy, are almost totally incapable of isolating the wanted signal from other transmissions and interference. Not surprisingly, the use of an audio filter ahead of such decoders will often make a critical difference and so enable the system to function correctly, rather than outputting gibberish.

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